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# Genetically Optimized Root Raised Cosine Filter in 3G Technology

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*Abstract*— **Wireless communication is the transfer of information over a distance without the use of electrical conductors or wires. In the last 50 years, wireless communications industry experienced drastic changes driven by many technology innovations. The application of signal processing to this area is becoming increasingly important. Indeed, it is the advances in signal processing technology that make most of today's wireless communications possible and hold the key to future services. The present paper deals with simulation model of raised cosine pulse shaping filter for WCDMA with different parameters of the filter at 5 MHz.**

*Keywords*- **FIR Filter, WCDMA, Simulation Model**

## I. INTRODUCTION

Wideband Code-division multiple access is one of several methods of multiplexing wireless users. In CDMA, users are multiplexed by distinct codes rather than by orthogonal frequency bands, as in frequency-division multiple access.

The enhancement in performance is obtained from a Direct Sequence Spread Spectrum (DSSS) signal through the processing gain and the coding gain can be used to enable many DSSS signals to occupy the same channel bandwidth, provided that each signal has its own pseudorandom (signature) sequence [3-4]. Thus enable several users to transmit their information over the same channel bandwidth. This is the main concept of a WCDMA communication system. The signal detection is accomplished at the receiver side by knowing the code sequence or signature of the desired user. Since the bandwidth of the code signal is chosen to be much larger than the bandwidth of the information-bearing signal, the encoding process enlarges or spreads the spectrum of the signal. Therefore, it is also known as spread spectrum modulation. The resulting signal is also called a spread-spectrum signal, and CDMA is often denoted as spread-spectrum multiple access. A trade-off exists between bandwidth containment in frequency domain and ripple attenuation in time domain. It is this trade-off of bandwidth containment versus ripple amplitude which must be considered by design engineers, when developing a data transmission system that employs pulse shaping.

The application of signal processing techniques to wireless communications is an emerging area that has recently achieved dramatic improvement in results and holds the potential for even greater results in the future as an increasing number of researchers from the signal process and communications areas participate in this expanding field. [3-6]. From an industrial viewpoint also, the advanced signal processing technology cannot only dramatically increase the wireless system capacity but can also improve the communication quality including the reduction of all types of interference. To satisfy the ever increasing demands for higher data rates as well as to allow more users to simultaneously access the network, interest has peaked in what has come to be known as WCDMA The WCDMA has emerged as the most widely adopted 3G air interface and its specification has been created in 3GPP. In this system the user information bits are spread over much wider bandwidth by multiplying the user data bits with quasi random bits called as chips derived from CDMA spreading codes. In order to support very high bit rates (up to 2 Mbps) the use of variable spreading factor and multimode connection is supported. The chip rate of 3.84Mcps/sec is used to lead a carrier bandwidth of 5Mhz.WCDMA also supports high user data rates and increased multipath diversity [5]. Here each user is allocated the frames of 10 ms duration during which the user data is kept constant though data capacity among users can change from frame to frame.

### A. *Pulse shaping filter*

In electronics and telecommunications, pulse shaping is the process of changing the waveform of transmitted pulses. Its purpose is to make the transmitted signal better suited to its purpose or the communication channel, typically by limiting the effective bandwidth of the transmission. By filtering

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the transmitted pulses this way, the inter-symbol interference caused by the channel can be kept in control.

Transmitting a signal at high modulation rate through a band-limited channel can create inter-symbol interference. As the modulation rate increases, the signal's bandwidth increases. When the signal's bandwidth becomes larger than the channel bandwidth, the channel starts to introduce distortion to the signal. This distortion is usually seen as inter-symbol interference.

The signal's spectrum is determined by the pulse shaping filter used by the transmitter. Usually the transmitted symbols are represented as a time sequence of Dirac delta pulses. This theoretical signal is then filtered with the pulse shaping filter, producing the transmitted signal. The spectrum of the transmission is thus determined by the filter. In many base band communication systems the pulse shaping filter is implicitly a boxcar filter.

Examples of pulse shaping filters that are commonly found in communication systems are:

- ✓ The trivial boxcar filter
- ✓ Sinc shaped filter
- ✓ Raised-cosine filter
- ✓ Gaussian filter

✓ *Raised Cosine Pulse*

In many data transmission applications, the transmitted signal must be restricted to a certain bandwidth. This can be due to system design constraints in such instances; the infinite bandwidth associated with a rectangular pulse is not acceptable. The bandwidth of the rectangular pulse can be limited, however, by forcing it to pass through a low-pass filter. The act of filtering the pulse causes its shape to change from purely rectangular to a smooth contour without sharp edges. [6] Therefore, the act of filtering rectangular data pulses is often referred to as pulse shaping. Unfortunately, limiting the bandwidth of the rectangular pulse necessarily introduces a damped oscillation. That is, the rectangular pulse exhibits nonzero amplitude only during the pulse interval, whereas the smoothed (or filtered) pulse exhibits ripples both before and after the pulse interval. At the receiver, the ripples can lead to incorrect decoding of the data, because the ripples associated with one pulse interfere with the pulses before and after it. However, the choice of a proper filter can yield the desired bandwidth reduction while maintaining a time domain shape that does not interfere with the decoding process of the receiver. [6] This filter is the well-known raised cosine filter and its frequency response is given by

$$H(w) = \tau \ldots\ldots\ldots\ldots\ldots\ldots\ldots .0 \le w \le c$$

$$\tau\left\{cos^2\left[\frac{\tau(w-c)}{4\alpha}\right]\right\}\ldots\ldots\ldots\ldots\ldots..c \le w \le d$$

$$0 \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots w > d$$

τ is the pulses period
α is Roll off factor
c is equal to π(1- α) τ
d is equal to π(1+α) τ

## II. Methodology

**Filter Design Techniques**

Here it is managed a portion of the procedures used to plan FIR filters: FIR filters are filters having an exchange capacity of a polynomial in z- and is an every one of the zero filter as in the zeroes in the z-plane focus the frequency response magnitude trademark. The Z transform of an N-point FIR filter is given by equation 1.

$$H(Z) = \sum_{n=0}^{N-1} h(n)z^{-n} \qquad (1)$$

FIR filters are particularly useful for applications where exact linear phase response is required. The FIR filter is generally implemented in a non-recursive way which guarantees a stable filter. FIR filter outline basically comprises of two sections.

- Approximation problem
- Realization problem

The approximation stage takes the specification and gives a transfer function through four steps:

- A craved or perfect reaction is picked, ordinarily in the frequency domain.
- A permitted class of filters is picked (e.g. the length N for a FIR filters).
- A measure of the nature of estimate is picked.
- A strategy or calculation is chosen to discover the best filter transfer function.

The acknowledgment part manages picking the structure to execute the transfer function which may be as circuit diagram or as a program. There are two well-known strategies for FIR filter outline specifically:

- The frequency sampling technique.
- The window method.

**The Frequency Sampling Technique**

In this method the desired frequency response is provided. The given frequency response is examined at a situated of similarly separated frequencies to get N tests. Thus, sampling the continuous frequency response $H_d(w)$ at Npoints essentially gives us the N-point DFT of $H_d(2\Pi nk/N)$. Thus by using the IDFT formula, the filter coefficients can be calculated as:

$$h(n) = \frac{1}{N}\sum_{n=0}^{N-1} H(K)e^{j(2\Pi n/N)k} \qquad (2)$$

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Presently utilizing the above N-point filter reaction, the consistent frequency response is figured as an addition of the inspected frequency response. The approximation error would then be precisely zero at the examining frequencies and would be limited in frequencies between them. The smoother the frequency response being approximated, the smaller will be the error of interpolation between the sample points. One way to reduce the error is to increase the number of frequency samples. The other way to improve the quality of approximation is to make a number of frequency sample specified as unconstrained variables. The values of these unconstrained variables are generally optimized by computer to minimize some simple function of the approximation error e.g. one might choose as unconstrained variables the frequency samples that lie in a transition band between two frequency bands in which the frequency response is specified e.g. in the band between the pass band and the stop band of a low pass filter. There are two different set of frequencies that can be used for taking the samples. One set of frequency samples are at $f_k$ = k/N where k = 0, 1….N-1. The other set of uniformly spaced frequency samples can be taken at $f_k$= (k + ½)/Nfor k = 0,1….N-1.The second set gives us the additional flexibility to specify the desired frequency response at a second conceivable set of frequencies. Thus a given band edge frequency may be closer to type-II frequency sampling point that to type-I in which case a type-II design would be used in optimization procedure. Rabiner and Gold [6], has specified a strategy focused around the thought of frequency sampling to outline FIR filter. The steps involved in this method suggested by Rabiner are as follows:

- The desired magnitude response is furnished alongside the quantity of examples. Excused N, the method decides how fine an insertion will be utilized?
- It was found by Rabiner that for designs, investigated, where N varied from 15 to 256, N samples of H(w) lead to reliable computations, so 16 to 1 interpolation was used.
- Given N values of $H_K$ , the unit sample response of filter to be designed, h(n) is deliberated utilizing the inverse FFT algorithm.
- In order to obtain values of the interpolated frequency response two procedures were suggested by Rabiner. They are:
  - h(n) is rotated by N/2 samples (N even) or (N-1)/2samples for Node to remove the sharp edges of impulse response, and then 15 N zero-valued samples are proportionally located about the impulse response.

h(n) is split around the N/2nd sample, and 15 N zero-valued samples are placed amid the two fragments of the impulse response. The zero augmented sequences are transformed using the FFT algorithm to give the interpolated frequency responses.

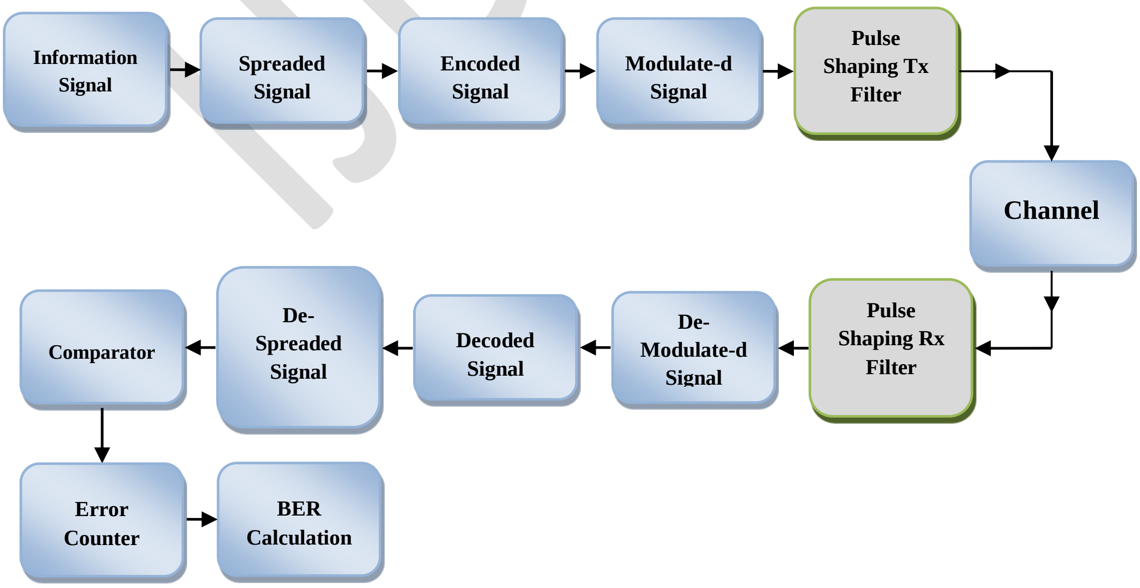


Figure 1: Block diagram for WCDMA System

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The WCDMA communication link is shown in Figure 1. The performance in terms of the Bit Error Rate can be examined for different values of Group Delay, D of the pulse shaping filter against a sinusoidal interference. A Simulink model based on the MATLAB provides the output. The information signal in wideband CDMA system is generated by Bernoulli Binary Generator and the PN sequence is used for spreading the signal at 5 MHz bandwidth. The signal is passed from different parameters block as shown in Figure 1 and at the end, BER is calculated by comparing the transmitted data and received data

**Genetic Algorithm**

Genetic algorithm is a part of evolutionary computing, which is a rapidly growing area of artificial intelligence. We can see that, genetic algorithm is inspired by Darwin's theory about evolution. Simply said, solution to a problem solved by genetic algorithm is evolved. In a genetic algorithm, a population of strings (called chromosomes or the genotype of the genome), which encode candidate solutions (called individuals, creatures, or phenotypes) to an optimization problem, is evolved toward better solutions. Traditionally, solutions are represented in binary as strings of 0s and 1s, but other encodings are also possible. The evolution usually starts from a population of randomly generated individuals and happens in generations. In each generation, the fitness of every individual in the population is evaluated, multiple individuals are stochastically selected from the current population (based on their fitness), and modified (recombined and possibly randomly mutated) to form a new population. The new population is then used in the next iteration of the algorithm. Commonly, the algorithm terminates when either a maximum number of generations has been produced, or a satisfactory fitness level has been reached for the population. If the algorithm has terminated due to a maximum number of generations, a satisfactory solution may or may not have been reached.

Genetic algorithms find application in bioinformatics, phylogenetics, computational science, engineering, economics, chemistry, manufacturing, mathematics, physics and other field. The steps in the typical GA for finding a solution to a problem are listed below:

1. Generate an initial solution population of a certain size randomly.
2. Calculate each solution in the current generation and assign it a fitness value.
3. Select "good" solutions based on fitness value and discard the rest.
4. If satisfactory solution(s) found in the current generation or maximum number of generations is exceeded then stop.
5. Change the solution population using crossover and mutation to create a new generation of solutions.
6. Go to step 2.

***Simulank model***

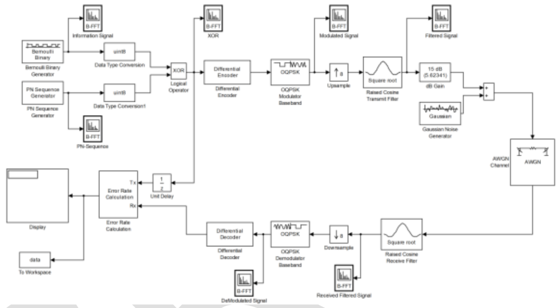

Figure 2: Simulink Model of WCDMA System

## III. RESULT

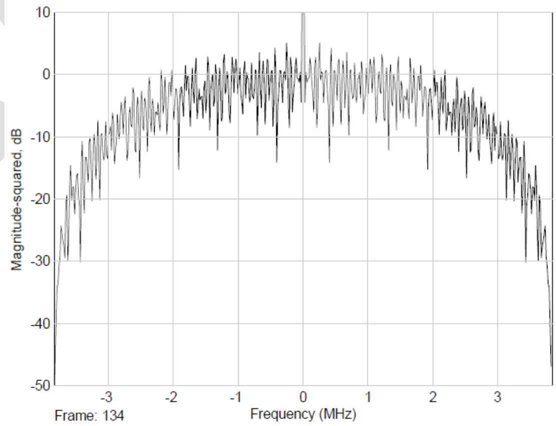


Figure 3: Modulated signal

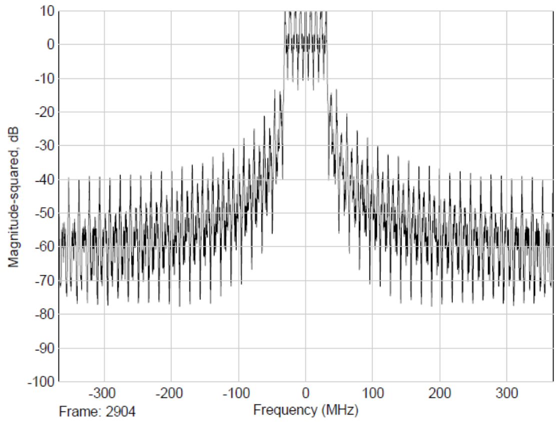


Figure 4: Transmitted signal

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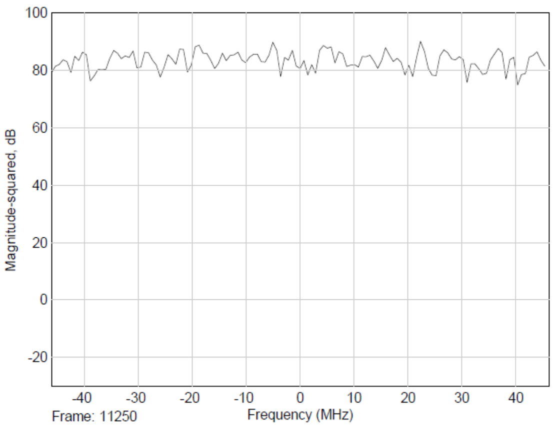


Figure 5: Received Filtered Signal

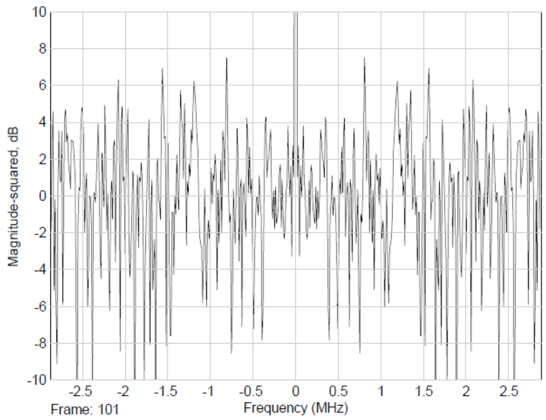


Figure 6: Demodulated signal

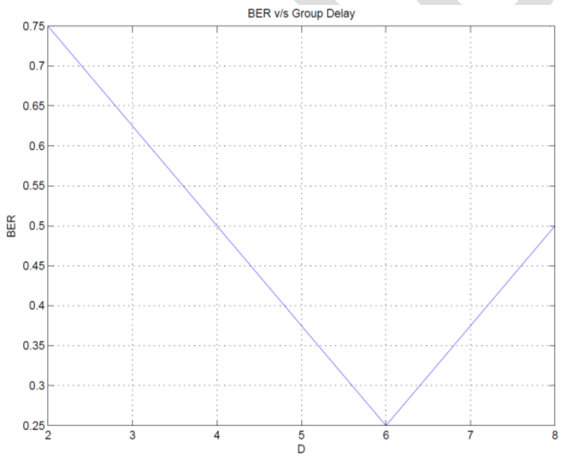


Figure 7: Effect of Group Delay on BER Performance

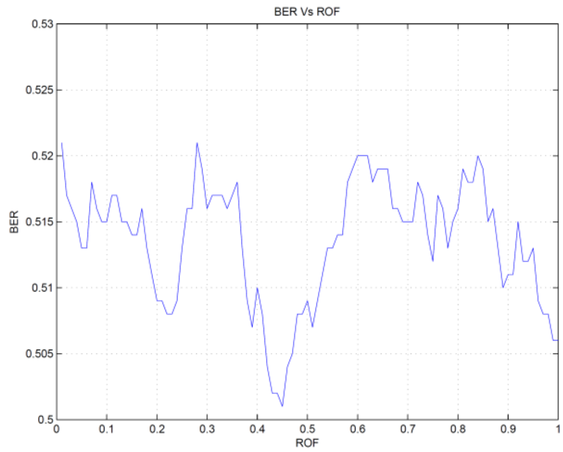


Figure 8: Effect of Roll off factor on Bit Error Rate

- ✓ previous work

Group delay = 6
Roll off factor = 0.22
BER = 0.502 over (3.0e+4 symbols)

- ✓ Our work

Group delay = 6 (by GA)
Roll off factor = 0.53 (by GA)
BER = 0.490over (3.0e+4 symbols)

We have shown the complexity analysis by this extension of work. The optimized parameters have improved the performance of system in terms of BER. The study is useful to improve the performance of WCDMA Network.

1. In the planning of WCDMA Network.
2. To achieve the flexibility in use of data rates in different environments.
3. Design of future cellular mobile communication network.
4. The proposed WCDMA Simulator can be used for optimization of parameters in various environments, with various mobile distributions and different services.

## IV. CONCLUSION

Square Root Raised Cosine filter has been analyzed for WCDMA at 5Mhz.The effect of variation of roll off factor, group delay and interpolation factor has been studied. It is necessary for RF design engineers to select the optimum value of D under the prevailing environment. The present study shows that opening of eye become more complex as the value of D is changed from 4 to 8 side lobe tail attenuation occurs more quickly as D is increased from 4 to 8. Hence the group delay must be controlled to decrease the complexity of the filter.

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Also we have shown the dependency of total BER over roll of factor. Future study may clear it more.

## REFERENCES

[1] J. Stevens, "DSPs in Communication," IEEE Spectrum Vol.35, No.9, pp. 39-46, Sept.98.

[2] C.Iniaco and D.Embres,""The DSP decision-fixed or floating?"IEEE Spectrum, Vol.3, No.9, pp.72-74,1996.

[3] R.M Piedra and A.Frish, "Digital Signal Processing comes of age" IEEE spectrum, Vol.33, No.5, May 1996.

[4] C. Richard Johnson and William A. Sethares, Telecommunications Breakdown.

[5] Simon Haykin, "Communications Systems, " Tata McGraw Hill.

[6] L. R. Rabiner, B. Gold and C.A. McGonegal, "An Approach to the Approximation Problem for Nonrecursive Digital Filters," IEEE Trans. Audio and Electro, vol. AU-18, pp. 83-105, June 1970.

[7] G.Giannkis,"Highlights of Signal Processing for communications"IEEE Signal Processing Magazine,vol 16,No.2,14-51,March,1999.

[8] Ken Gentile,"The care and feeding of digital pulse shaping filters"www.rfdesign.com,pp-50-61, 2002.

[9] John G. Proakis and Masoud Salehi, Communications Systems Engineering.

[10] B.P. Lathi, Modern Digital and analog Communications Systems, John Wiley & Sins.